

1/PRTS

09831596 002201

JC08 Rec'd PCT/PTO

11 MAY 2001

PCT/GB99/03753

WO 00/28521

Audio Dynamic Control Effects Synthesiser with or without Analyser

Patent Application PCT GB97/02159 (WO98/0714) ("The Prior Application") describes an audio effects synthesiser with or without analyser and should be read  
5 along with this description of further improvements.

The purpose of this invention is to store the characteristics of audio level control devices which have audibly desirable properties and to be able to synthesise these properties at will.

10

Summary of Figures

- Figure 1 shows an analysis arrangement for a device under test (D.U.T.).
- Figure 2 illustrates two characteristics of a compressor device namely the gain  
15 characteristic and the time dependent characteristic.
- Figure 3 shows a flow diagram of the process of analysing the gain characteristic of a D.U.T.
- Figure 4 shows a table of derived figures from a hypothetical 'ideal' compressor and some example figures which may be obtained in reality.
- 20 Figure 5 shows graphically the values of figure 4.
- Figure 6 illustrates the derivation of intermediate ratio curves from measured curves.
- Figure 7 shows the application of test impulses as required by The Prior Application in conjunction with a gain varying device.
- Figure 8 shows a flow diagram for assessing the time constant characteristic of a  
25 D.U.T.
- Figure 9 shows an arrangement for applying the simulation of a gain varying device in conjunction with the simulation of non-linearity described in The Prior Application.
- Figure 10 shows a flow diagram of the implementation of the time constant characteristics of a device during simulation.
- 30 Figure 11 shows an example of an alternative way of collecting a set of impulse response data to simulate novel compression effects.

Figure 12 shows another example of an alternative way of collecting sets of impulse response data to further simulate compression effects.

Figure 13 shows in more detail the bi-linear interpolation method.

### 5 Characteristics of a Dynamic Range controlling Device.

Figure 1 shows a typical arrangement of an audio compressor device 1 (in this case the device under test), being driven by a signal generating arrangement 2, and being analysed by the signal analysis arrangement 3.

10

A compressor is designed to reduce the dynamic range of audio program material by reducing the amplitude of louder passages in relation to that of quieter passages. In a typical compressor as shown at 1, the input signal is fed (often with some pre-  
15 amplification and or variable attenuation) to a gain control element 11. This alters the amplitude of the signal which is then fed again usually via some amplification or buffering to the output where there is usually some additional variable gain available to compensate for attenuation in the overall signal path.

20

Some of the output signal is usually fed to a rectification circuit where controls 12 and 13 select a threshold above which gain reduction is to be effected and an amount or ratio by which gain is to be reduced according to the output signal. The rectified signal is further applied to a time constant arrangement where controls 14 and 15  
25 apply attack and recovery time constants by which gain is reduced and restored when the output signal increases or decreases.

25

Note that there are a number of variations possible in this arrangement and also it is possible to rearrange the gain control circuitry so that dynamic range expansion is produced or so that signal gating may occur. These do not affect the operation of the analysis device.

30

The operation of the device under test is determined by the signal generating arrangement 2 comprising the choice of a continuous signal generator 21 or a pulse

3

generator 22 fed to a variable gain control element 23 and then to the device under test via a digital to analogue converter 24.

Note that the D-A converter may be omitted if the device under test is actually a digital device or if the signal generating arrangement is implemented in analogue circuitry. For the rest of this discussion the assumption is that the device under test is analogue and the analysis and synthesis device is digital.

The output of the device under test is fed to the signal analysis arrangement 3 comprising an analogue to digital converter 31 and the choice of level detection circuitry 32 and impulse storage circuitry 33.

The entire analysis arrangement is controlled by system controller 34 which is typically a data processing arrangement which may share processing hardware with various elements of the complete analysis and synthesis system.

Figure 2 shows the main characteristics of the device under test to be assessed.

Fig 2 (a) shows the gain characteristic of a typical compressor. Curve 41 shows the relationship between the input level of a signal fed to the device and the resulting output at a particular setting of the controls. It can be seen that at low levels the output rises linearly in proportion to the input, along the dotted line 43 such that a 1dB increase in input results in a 1dB increase in output.

As the input level approaches the threshold setting 42 the gain begins to modify to approach that of line 44 which represents the ratio or slope of the compressor. This may be for example that it requires a 3 dB increase in input to produce a 1 dB increase in output (a ratio of 3:1). There is usually no sudden transition at the threshold and the degree of 'softness' of the knee of the curve is one distinguishing characteristic of a particular compressor. It is also possible to have multiple thresholds between multiple ratios.

Fig 2 (b) shows the time constant characteristic of a typical compressor. The input signal is shown as a continuous sine wave which doubles in amplitude (increases by 6dB) at 45 and then decreases by 6dB at 46. The corresponding output shows that there is an initial increase by 6dB at 47 following which the gain is reduced according to the attack time constant until it reaches an essentially stable reduced gain at 49 determined from the curve of part (a). When the input signal reduces by 6dB the output also reduces immediately by 6dB at 48 then the gain is increased according to the release time constant until the appropriate gain for the input as determined by the curve in part (a) is restored by point 50.

10

Time constants are typically operator variable and the range and options provided by a particular device contribute to the user perceived operability of the device and to the audible characteristics of the device under test. However it is not essential to measure these characteristics as a standard set of time constants can be provided which are common to most devices to be simulated. In the event that it is desired to assess this characteristic of a device under test to more closely simulate a specific device the method is described below.

15

The third characteristic that determines the audible effect of a gain control device is the distortion or non-linearity caused by the gain control element. This may be assessed using the means of The Prior Application with some small modifications as set out below.

20

#### Analysis of the gain characteristic.

25

To properly analyse a device under test to determine its gain characteristic it is necessary to take at least one set of measurements corresponding to a specific setting of the ratio control on the device. As is described later the characteristic of other ratios may be interpolated between any pair of measured ratio characteristics and it is also possible to further interpolate between a measured ratio characteristic and an ideal 1:1 (no compression) and infinity:1 (limiting compression) characteristic.

30

In this example three sets of measurements are taken, for example for a ratios of 1.5:1, 5:1 and 20:1 (although any value or values spanning commonly used ratios may be chosen). To take each set of measurements the attack and decay settings of the device should each be set to either their maximum settings or to 1 second, whichever is  
5 shorter.

To take the set of measurements corresponding to the ratio of 1.5:1, this setting is selected on the device under test, and if a threshold control is available this is set to -20dB, and the signal generator of figure 1 is set to generate a 1kHz sine wave tone  
10 (although other frequencies may be used as described later) at a typical operating level for the device, chosen to be suitably below the overload limit of the device. For a professional unit a level of 0dBm is appropriate. If available any make-up or output gain of the device under test should now be adjusted so that the level returned to the level detector of figure 1 is also at 0dBm once the gain of the device under test has  
15 reached a steady state. If such an output adjustment is not available and this level is not achievable it is permissible to use any appropriate output level from the device.

Figure 3 shows a flow diagram of the test procedure which the signal generator and level detector conduct under computer control once the above conditions are set up. A  
20 sequence of measurements are taken for signals fed to the device under test between -40dBm and 0dBm at 1 dB intervals. As each level generated is set up, the output level from the device under test is repeatedly assessed until this level becomes stable to within 0.1dB over a 100mS interval. The output level is then recorded for this input level.

25

The amplitude interval with which the tests are conducted is dependent on the accuracy required in the simulations. The 1dB steps described give good results for most applications but if there are time constraints on taking the tests optionally larger steps may be used and data may be interpolated to make up for missing detail.

30

The set of device output levels corresponding to each input levels step is recorded within the memory of the computer and may be represented by the first column of table (a) of figure 4.

- 5 The above set of tests may be repeated for as many ratio settings as desired, for example for ratios of 5:1 and 20:1, shown in the second and third columns of table (a) of figure 4.

The number of different sets of measurements each corresponding to a ratio setting to  
10 be taken depend on the degree of accuracy to which it is required to simulate the device under test and also the number of options available on the device. Simple devices may only have one or two slope options in which case these provide sufficient data for storing the characteristics of this device. The later description of the use of this data shows how available data is interpolated to provide options sometimes  
15 beyond what was available with the device under test.

Figure 4 (a) shows an example table of data derived from the device under test with some entries omitted for clarity. In fact these figures would be derived from a device with a 'perfect' gain characteristic as can be seen from the plot of these figures in  
20 figure 5 (a) which shows that there is no 'soft knee' as would be encountered in reality. Such a device may not sound ideal but serves to illustrate the process more clearly. It should also be clear that the ratio setting of the device means the change in output level divided by change in input level expressed in dB provided that the signal is well above the threshold.

25

(ratio = (delta dB in) / (delta dB out)).

If the ratio settings are not specified on the device under test the analyser may assess the curve by inspection of the data at the higher levels applied and assign a value to  
30 the ratio based on the above formula. It is also possible to store the actual text used to describe the ratios on the device under test (provided the operator enters this data) and

provide these options during simulation rather than the derived figures as an optional closer match to the user interface of the device under test.

These figures also show an exact setting of threshold at -20dBm at the input to the device. In practice if this varies with ratio settings the figures will not be so clear to interpret but the following processing still applies and the variations in the device will remain embedded within the data for simulation. If a calibrated threshold setting is provided but -20dB is not available, it is acceptable to use another convenient threshold setting and ideally the analyser will be arranged to allow this value to stored along with the analysis data. This threshold figure may then be used during simulation along with the operator's desired threshold to achieve an accurate simulation by adjustment of the values as described below.

Figure 4 (b) shows some more typical data which may be obtained from a real device and this serves to illustrate the next stage of processing.

Although figures are obtained for an input range of 40dB these figures are extrapolated by determining the actual slope at the upper end of the graph by averaging over the 3 or 4 highest measurements and using this slope to generate a notional straight line extrapolation at any further higher input level. At the lower end of the graph a slope of unity is assumed and this is used to extrapolate down to arbitrary low levels of input.

#### Simulation of the gain characteristic

25

During simulation of the gain characteristic of the device which was under test, the operator must first select a desired ratio  $R$  of compression to be applied to the signal to be processed by the simulation.

30 As the input signal is processed the amplitude is continually assessed according to details to be described. The input level is thus determined to be say  $I$  dB and this is indicated in figure 5(b).

8

If the value R matches the ratio of one of the stored curves, say 5:1, the desired output level may be immediately assessed by inspection of the data, and if the value I falls between two stored input values (say I1 and I2 in the diagram) a simple linear interpolation may be used to determine the desired output P. From the known input  
5 and desired output an attenuation G (which may be negative, i.e. an attenuation) may be simply calculated as

$$G = P - I,$$

10 and applied to the input signal as described below.

If the value R does not match the ratio of one of the stored curves but there is data for ratios greater and less than the value R then the desired output level can be determined by interpolation between the two stored ratios as follows with reference to figure 6.

15

Figure 6 shows a graphical representation of typical stored data where the curve for 5:1 is known (curve 73) and also that for 20:1 (curve 72). It is desired to interpolate curve 71 representing a ratio of 10:1 from this data. Also shown are ideal curves 1:1 (straight line 75) and infinity:1 (curve 74 which has a sharp knee where it meets curve  
20 75 and progresses down at unity gain ratio along line 75).

The curves shown incorporate various approximations inherent in the device originally tested to produce this data including errors in ratio, differences in threshold and small offsets in gain at low level as would be expected from real devices. The  
25 interpolation described reflects these imperfections and provides a smooth transition from one curve to another under user control. Any interpolation towards one of the ideal curves will again produce a smooth transition from the measured curve to the ideal curve.

30 The following describes in detail the interpolation steps..



If curves are known for the higher ratio of M:1 and the lower ratio of N:1, and we desire to know the required output level at a given input level with a ratio of R:1, first determine the coefficient of linear interpolation  $a$  from the formula

$$a = N(M-R) / (R(M-N)). \quad (1)$$

Then if the output level at the higher ratio M:1 would be X dBm, and the output level at the lower ratio would be Y dBm determined from the relevant curves (using linear interpolation if necessary as described above), the output level Z we require for ratio R is

$$Z = (1-a)X + aY. \quad (2)$$

Therefore, first considering the example based on ideal curves where M=20 and N=5 are known curves (72 and 73) and it is desired to determine the output for a curve of R=10 (curve 71), substituting in equation 1,  $a$  is determined to be as follows:

$$a = N(M-R) / (R(M-N)) = 5(20-10) / (10(20-5)) = 5 \times 10 / (10 \times 15) = 1/3.$$

Referring again to figure 6 the dotted curve 71 representing a ratio of 10:1 may thus be derived by using the  $a = 1/3$  at any input level to perform a linear interpolation between the two curves 72 and 73 according to formula (2) where X is the output level from curve 72 and Y is the output level from curve 73.

Note that although this interpolated curve may not be identical to a measured curve for this ratio from a device under test, the interpolated value satisfies the two criteria that (i) it does represent a reasonable derivation of the unmeasured value and as R is varied between the ratios of the two curves the interpolated curve smoothly transforms from one of the extreme curves to the other, and (ii) in the case of idealised curves the method produces an exact solution.

10

If the value R does not lie between two stored ratios one approach is to deny the setting to the user, offering the nearest alternative stored ratio as it was not possible to determine what the device under test would have provided for ratio R. Alternatively a desired ratio may be derived by interpolation using the above method between the nearest known ratio and (for a larger ratio) the ideal curve for an infinite ratio, as shown at 74 in figure 6, and (for a smaller ratio) a unity gain curve representing a 1:1 ratio, as shown at 75 in figure 6. In the case of interpolating to an infinite ratio, the formula (1) needs to be adjusted as follows:

$$a = N/R. \quad (3)$$

Thus a is unity if R is chosen to equal N as expected and a falls to zero as R tends towards infinity.

Where interpolation is towards a 1:1 ratio, equation 1 also simplifies with N=1 to

$$a = (M-R)/(R(M-1)). \quad (4)$$

Thus when R is equal to M, a is zero, so the curve tends to that of M, and when R is 1, a is 1, thus the curve becomes exactly the 1:1 curve and no compression effect results.

The interpolation formulae (2) may also be applied directly to gains derived from each ratio rather than absolute output level for any given input level as the formula is independent of any constant added to each term, X, Y and Z.

25

#### Determination of Time Dependent Characteristics

In order to determine the time constant characteristic of the device under test it must be analysed for dynamic response. It is possible to take a number of test results from the device at different ratio settings if it is desired to simulate the device at greatest accuracy but generally the same time constants apply at all ratios and it is possible to make the measurements at one fairly high ratio, for example 20:1.

The device under test should thus be set to a 20:1 ratio with a threshold of -20dBm or as close to this as possible and the signal generator 2 of figure 1 is set to produce a 0dBm tone at typically 1kHz. The output or make-up gain should be adjusted for a -20dB output from the device under test or as close as practical so that overshoots during test do not cause distortion.

The controls are first set to the fastest attack and release times available on the device under test, and then the analysis process is started and proceeds automatically as indicated in figure 8.

10

It is assumed that the output of the test generator is already at 0dBm into the device under test. The test output is first reduced to -20dBm (step 80) and the output from the device is repeatedly assessed (step 81) in the level detector 3 of figure 1 until the level has stabilised to a variation of less than 0.1dB over the period of typically 1 second.

15 The variation in output level from the device is stored repeatedly during this process to determine the dynamic characteristic of the recovery time constant of the device.

The generator output is now increased again to 0dBm (step 82) and again the output of the device is repeatedly assessed and stored (step 83) until the output is again stable to within 0.1dB per second.

20

A number of ways of determining level during a rapidly changing signal are possible but since the signal generator and level detector are under full control of the system controller one method with fast response is to arrange that the frequency of the test tone is synchronous with the sampling rate of the generator/detector combination. Since the measured waveform is now synchronous with the sampling rate, the level calculation needs only be performed for one cycle period, repeatedly until stable.

25

From these measurements the time constant characteristic for the fast setting of the device under test can be determined and stored. All the gain values for this derivation are considered as multiplicative factors, not as logarithmic gain changes. The time constant is defined as follows: During a change in gain from G1 to G2 it is the time taken for

30

12

the gain to reach  $G_3$ , where  $(G_3 - G_2) / (G_1 - G_2) = 1/e$ , where  $e$  is the exponential constant. Since the initial and final gain is known the period to reach this gain may be calculated from measured characteristic. This derivation applies when the input signal is increased suddenly in level. When the input signal is reduced suddenly, the gain recovery is usually exponential towards the gain setting applicable for zero input signal, so  $G_2$  should be taken as this limiting gain and if the gain does not lie between  $G_1$  and  $G_2$  under these conditions the time constant should be derived from a smaller change in gain according to the assumption that the gain follows a similar exponential curve.

10

It is also possible that a multi-stage time constant circuit is included, and inspection of the amplitude change characteristic can determine this for use in simulation.

Many existing devices offer a limited set of attack and recovery time constants, often labelled as fast/medium/slow and thus the measurements above should be repeated for each of these settings. In simulation the operator can be offered these simple choices which can recall the stored time constants when selected. In addition the user can be offered a continuously variable time constant to extend the simulation beyond that achievable with the original device under test, and where all three speed options are not available preset defaults can be supplied.

20

#### Determination of Distortion Characteristic at One or Multiple Attenuation Levels.

The final characteristic of the device to materially affect the audible performance is the non-linearity of the gain control element. Due to the analogue nature of many such gain elements significant non-linearities may be present both due to the device itself and due to the feedback of any audio signal into the gain control signal 16 of fig 1. The non-linearity of the device may be assessed in the following manner using and extending the techniques of The Prior Application.

30

There are three approaches to analysing and simulating the non-linearity of the gain element. One is to determine a single characteristic set of impulse responses according

- to the method of The Prior Application of the gain control element in the linear portion of the curve of the device under test. This may be performed by setting the threshold control of the device sufficiently high that the test sequence of impulses of the The Prior Application do not at any time cause any gain reduction to take place.
- 5 In addition if the device has a 1:1 setting this should be used to prevent gain variation. This establishes a fixed analysis of the of the gain control device which may be used in the simulation as a fixed processing characteristic to be applied to an input signal in addition to the gain control adjustment already described.
- 10 A further method is to assess two characteristic sets of impulse responses of the gain control at two different attenuations. Since most gain control devices operate in the range of zero gain reduction to 20dB of gain reduction, it is desirable to take two measurements of the distortion of the gain control device at these two extremes and to linearly interpolate them during simulation using methods of bilinear interpolation as
- 15 disclosed in The Prior Application and shown in figure 13, in which one dimension of interpolation is the instantaneous amplitude of the incoming signal being interpolated between adjacent impulse levels, and the second dimension is interpolation between non-linearities of the two gain characteristics according to the degree of attenuation to be applied.
- 20 A third method is to take multiple assessments of the distortion characteristic at multiple gain reductions and to pair-wise interpolate to get the desired characteristic for any given required gain reduction.
- 25 In order to take measurements of the gain control element at specific attenuations it is necessary to modify the test signal described in The Prior Application to ensure that each test impulse is applied at a known attenuation of the device under test.
- In order to do this the device under test should be set to slow attack and decay
- 30 characteristics and a large ratio, say 20:1 if available.

Using the techniques already discussed to assess the gain characteristic it is now possible to apply a sine wave tone, typically at 1kHz, or at a frequency close to this that this synchronous with the sampling rate, that results in a known attenuation. This level is then maintained until the gain of the device under test has stabilised, again  
5 using the techniques already described.

Figure 7 (a) shows the sine wave being applied at 91 to achieve the desired gain. The sine wave is then removed (over short period T1 to minimise the impulsive effect). A period T2 is then allowed to elapse to allow any stored energy to dissipate. Figure 7(b)  
10 shows the output of the device under test which is being analysed by both the level detector 32 of figure 1 and the impulse storage unit 33. The signal 95 has been measured to indicate a steady state at the desired gain reduction. During period T2 any ripples 96 are ignored.

15 Step impulse 92 is applied after T2 and held as the response to this impulse 97 is stored. Finally after period T3 the step impulse is removed and the sine wave restored. The recovered wave 99 is then allowed to stabilise before the process is repeated for the next different amplitude impulse.

20 In this way a complete set of impulses are sequentially applied to the device in a known stable gain state and stored and processed according to the method of The Prior Application. In this way a characterisation of the non-linear response of the gain control element at the know gain reduction is obtained.

25 The sequence may be applied at a different desired gain to assess the non-linear response of the device at this different gain. In this way it is possible to store either two response sets for the two desired gains at either extreme of the range or if desired a complete set of responses spanning the desired range which may be pair-wise interpolated during simulation.

30

In experiment it has been determined that gain control elements demonstrate comparatively short impulse responses and so the entire impulse response may be

captured before the gain has significantly changed. In practice it is found a settling time T2 of about 100 samples at 48kHz is sufficient and a storage time T3 of 200 samples is sufficient to encapsulate the performance of the device under test. Thus the test is completed for each impulse in under 10mS from the discontinuation of the level  
5 setting tone 91. Recovery time constants of about 1 S are therefore sufficient to hold the gain sufficiently constant during this period.

It should be noted that if the make-up gain controls are not adjusted during taking a sequence of measurements (as would be expected during automated testing) and as the  
10 absolute levels of the impulse response are recorded, the gain reduction as well as the non-linearity characteristic is encapsulated in the stored impulse response.

The simulation process (described in detail below) applies an appropriate impulse response (or set of responses according to the Prior Application) dependent on the  
15 gain reduction to be achieved in the simulation. In addition it is capable of applying additional attenuation when simulating desired gain reductions outside the range sampled impulse responses with attenuation inherent in them. An exactly equivalent final result can be achieved in two ways, the choice being dependent on the implementation details of the system, for example, the amount of real time computing  
20 power available during the simulation.

Method 1 is to leave the impulse response (or sets of them) as sampled, with their inherent attenuation. In this way as the correct response is automatically selected the correct attenuation will be achieved. Intermediate attenuations can be achieved by an  
25 appropriate choice of linear interpolation coefficient. Extrapolated attenuations are achieved in simulation by using additional gain modification by means of a digital multiplier.

Method 2 is to eliminate the inherent attenuation from the sampled impulse responses.  
30 This process is known as normalisation. Since the exact attenuation inherent in each impulse-response (or set of them) is known according to the method described above, it is possible to multiply every element of the impulse response (or set of them) by a

constant which is the inverse of this attenuation factor. In this way, during the simulation, the impulse response selection and interpolation are used solely to determine the impulse response characteristic appropriate to an attenuation. The chosen impulse response will then not result in any intrinsic attenuation. The  
5 appropriate gain reduction is then applied independently in the additional gain reduction element (113 in example simulator to be described). This method allows the audio processor to offer the choice of simulating the gain reduction characteristic (in terms of input level to output level) with or without the simulation of the gain reduction signal quality sampled from the device under test, or with a user selectable  
10 limitation on the range of impulse responses to be applied.

### Simulation of Level Control Device

Figure 9 shows an overall diagram of the simulator of level control devices.

15

Analogue to digital converter 101, which may be omitted if the input signal is already in digital form, takes an input analogue signal and feeds the signal to the modified convolution processor device 102 as described in The Prior Application comprising the modified convolution device 103 which convolves the incoming signal with the  
20 impulse response stored in the Finite Impulse Response Set (FIR set) storage device 105, under control of the amplitude assessment device 104 which selects on a sample by sample basis an appropriate response or pair of responses from FIR set storage 105 and also supplies an interpolation value 120 to processor 103.

25 The convolution processor 102 may also contain additional inputs 121 and 124 which receive a second interpolation value 121 (if required) and FIR set selector value 124 (if required), this FIR set selector value selecting between FIR sets representing different gain reductions of the original device under test, and the interpolation value 121 providing interpolation values between them.

30

Address value 122 is used to select the appropriate offset in each FIR appropriate to the convolution operation and the amplitude assessment unit provides up to two FIRs



from each set providing two values for interpolation, with set selection 124 selecting two sets of values to use, thus providing 4 data values on input 125 to the convolution processor for each step in the convolution. The two interpolation values 120 and 121 provide for bilinear interpolation between these four values selected on a sample by  
5 sample basis of the input signal.

The input signal is further applied to the amplitude assessor with time constants 106 which determines an envelope for the input signal on a sample by sample basis under control of the user selected time constants 109. The resulting envelope 126 is fed to  
10 gain reduction calculation device 127.

Adjustment is first made according to any user desired change in the threshold setting. If the original measurements were taken with an assumed threshold of -20dB, and the user now select a threshold of -10dB, the difference (10dB) is subtracted from the  
15 input level derived in order to determine the gain variation required to simulate by reference to the derived gain characteristic data. In general if the original measurements of the gain characteristic were taken at a threshold of -A dB and the user now desires a threshold of -B dB the amount (A-B) dB is subtracted from the derived input signal envelope.

20

Gain table data according to figures 4 and 5 is stored in Gain Table Memory 128 and whenever the user selects a new ratio a table appropriate to the desired ratio is calculated according to the description above in reference to figure 6 and stored in memory 130.

25

The envelope of the input signal, modified by a user change in desired threshold, is fed to memory 130 to determine the desired output level for the signal of this envelope, and accordingly to select a desired gain for the signal at this instant.

30

The desired gain is fed to impulse set selector 107 which selects the desired FIR or set of them according to the method to be described below (under the heading "Gain Reduction Processor") and derives signal 124 which is fed to the impulse response

memory addressing, and (in the case where a pair of impulse responses or sets of them is used) a linear interpolation value signal 121 which is fed to the convolution processor.

5 Where the gain required is embodied within the FIR (or set of them), no additional gain adjustment is required (except for any user specified makeup gain described below). Where the gain required is not embodied in the FIR (or set of them) a gain adjustment signal 108 is derived and fed to the additional gain control element 113 via gain combiner 112.

10

User required make-up or overall gain may be applied by user selection 111 providing an extra gain demand signal which is combined in combiner 112 (which may be an adder if the gain signals are in logarithmic form, for example specified in dB, or a multiplier if specified in the form of a multiplicative value). The resultant gain signal  
 15 is fed to the additional gain element 113 which applies this gain to the output of the convolution device and feeds the resultant signal (by way of digital to analogue converter 114 if desired) to the output of the device.

#### Amplitude assessment with time constants

20

Amplitude assessment may be performed in a number of ways familiar to a skilled person. An example is shown in figure 10. The absolute value of an input sample S is determined and is known as |S|. This is compared with the current envelope E. If it is greater, the value E is increased according to the amount |S| exceeds E under control  
 25 of the attack constant k1, as follows

$$E = E + k1 \cdot (|S| - E).$$

The release constant is then applied which results in an exponential reduction in E:

30

$$E = k2 \cdot E.$$

It is possible to elaborate this with two stages of time constant or with additional low pass filtering to remove unwanted audio modulation of the derived envelope with improvements in overall distortion characteristics but the above is sufficient for an acceptable simulation.

5

The constants  $k_1$  and  $k_2$  may be related to analogue time constants  $T_1$  and  $T_2$  by noting that  $1 - k_1$  raised to the value  $r$  gives  $\exp(-1/T_1)$  and  $k_2$  raised to the value  $r$  gives  $\exp(-1/T_2)$  where  $r$  is the sample rate in samples per second.

#### 10 Gain Reduction Processor

By reference to the gain characteristic of the device under test an appropriate gain table is derived for the ratio desired by the operator according to the method already described. This may be calculated once each time the user selects a desired ratio  
15 setting.

On a sample by sample basis the envelope signal derived from the amplitude assessment of the input signal is determined and converted to logarithmic form (i.e in dB relative to 0dBm input). If at this stage it is desired to adjust the threshold from  
20 that used during the sample, an increase in desired threshold is subtracted from the envelope (a decrease is added) and the result used to determine a desired gain from the table of output amplitudes versus input amplitudes. If the value falls between table entries a simple linear interpolation is applied.

25 Once the gain required is determined, it is now necessary to determine whether there is a FIR set stored appropriate to this gain value. If there is then this FIR set is selected into the convolution selection algorithm. If there is not a FIR set for this gain reduction, but there is for two other gain reductions on either side of the required gain, these two FIR sets are selected into the convolution algorithm and an interpolation  
30 value is generated to achieve the desired intermediate effect. In the case where the attenuations are embodied in the FIR (or sets of them) the interpolation factor may be derived as follows.

20

If FIR set A gives an attenuation of a dB and FIR set B gives an attenuation of b dB, then the multiplicative factor appropriate to each Ma and Mb is as follows

$$Ma = 10^{(a/20)}, \text{ and } Mb = 10^{(b/20)}.$$

5

Applying an interpolation factor of j to FIR set A gives a total multiplicative gain Mg of

$$Mg = j.Ma + (1-j).Mb.$$

10

And the desired gain in dB  $G = 20 \log(Mg)$ , or  $Mg = 10^{(G/20)}$ .

Thus j may be derived for this sample from the rearrangement

$$15 \quad j = (Mg - Mb) / (Ma - Mb). \quad (5)$$

Ma and Mb are known in advance for each FIR set so it is only necessary to calculate Mg and hence j for each sample. In addition a list may be kept at for example 0.1dB intervals of all desired gains and the resulting choices of FIR sets and interpolating  
20 factors so that simple table look up may be performed on a sample by sample basis.

In the situation where the desired gain is greater than or less than that for every stored FIR set. The single FIR set embodying a gain closest to the desired gain is selected into the convolution algorithm and an additional gain factor F is generated such that

25

$$F = G - A,$$

where G is the desired gain at this instant and A is the gain of the nearest FIR set. The additional gain F in dB is added to any make up gain in 112 and passed to the  
30 additional gain element 113 to generate the desired overall gain.

In the case where the FIRs (or sets of them) do not have attenuations embodied in them, i.e., they are normalised to a fixed gain, the interpolation factor can be derived as described above to generate the desired audible signal processing characteristic although the convolution processor does not provide any gain adjustment to the  
5 signal. In this case the entire gain adjustment value  $G$  is fed to combiner 112 to implement the gain adjustment characteristic of the simulation. It should be noted that this allows the interpolation factor  $j$  to be derived in other ways, such as simple linear interpolation of the desired gain expressed in dBs between the gain values in dBs appropriate to the pair of FIR sets. Both the method described and this linear  
10 interpolation provide acceptable results and the choice will be open to the system designer according to available processing power.

#### EQ in the side chain

15 Where the system is used for sampling a complete audio processing channel including a gain control element, there may be some equalisation applied between the input and the gain controlling device. This results in some input signals being dynamically adjusted differently from others depending on frequency spectrum. A similar effect is achieved if a gain controlling device provides EQ in the side-chain (i.e. between the  
20 main audio signal path and the level detection section of the gain controlling device 1 of figure 1.

Both these variations can be handled by taking an additional analysis of the signal path between its input and the gain controlling device by taking a single impulse  
25 response test of this path to determine the frequency characteristic. This characteristic may be replicated in the optional additional EQ unit 129 of figure 9.

This additional equalisation is not usually critical to the operation of the overall system and it may be implemented by a straight finite impulse response convolution  
30 with the data derived from the above test or may be simulated by a derived infinite impulse response equaliser of broadly similar frequency response.

The existence of equalisation prior to the gain control element or in the side chain may suggest alternative frequencies for use in the gain reduction analysis such that the test signal is passed properly into the gain determining elements. It is also possible to use a number of different test frequencies to analyse the behaviour of the signal path prior to the compressor without reconnecting the test arrangement of figure 1 for subsequent synthesis.

Use without the Non-Linear effects synthesis of The Prior Application

Although this system has been described using the non-linear effects analysis and synthesis of The Prior Application it is possible to determine a single impulse response at each gain setting measured to determine frequency response and relative gain and apply the above technique without the non-linear section of the algorithm. In this way a restricted but useful synthesis of the device is achieved simulating the gain characteristic, the frequency response and variation with gain reduction (if multiple gain reductions are sampled) and also simulations of time constant if implemented. This may even be desirable if it is determined that the non-linearity of the original device is not desirable in a given simulation so in any event the degree of non-linearity should be made variable as described in The Prior Application and also to include using a single impulse response from a derived set of data.

It is also possible to omit the impulse response determination completely and to use the methods of the invention to simulate the gain characteristic alone or in combination with the time constant characteristic. In such a system, the table of input levels to output levels as for example shown in figure 4 is derived from a gain controlling device and stored for future simulation. It is then possible to simulate the gain reduction characteristic without the convolution achieving at least that part of the desirable characteristic of the original device embodied in this transfer curve. Such a simulation system, with reference to figure 9, would omit the convolution processor 102 and the selector circuitry 107.

Use without Analvser or with data derived from alternate uses of the analyser

Although the description explains how to take measurements of existing audio dynamics processor devices and how to simulate them, it is also possible to generate  
5 the necessary data for synthesis by means of any desired model of such a processor or by mixing data obtained from different sources.

In particular:

- 10 1. The curves of gain characteristic can be calculated for a computer model of a desired effects device, or may be drawn by an operator on a computer screen either completely freely, or may be modified manually by graphical manipulation from existing data. In any of these ways an operator may obtain a desired gain characteristic not available in a real dynamics processor device.
- 15 2. Time constant characteristics may be freely substituted for any measured characteristics, or derived by operator choice to more closely suit the audio material to be processed.
- 20 3. Characteristic FIR sets (either linear or non-linear) may be generated by computer model of a gain control device that it is required to simulate, for example there are now a number of ways to simulate valve circuits, and it would be possible to generate arithmetically the impulse response of these simulations appropriate to a variety of levels of impulse and gain variations of the simulated device. These impulse  
25 responses may then be used with the methods described herein to process audio according to these derived characteristics.
- 30 4. Characteristic FIR sets (either linear or non-linear) may also be derived from other audio devices or systems which have a desirable characteristic and used with the methods described herein to simulate a dynamic control device not currently achievable.

One example is illustrated by reference to figure 11 in which it is shown how to derive two or more sets of impulse response characteristics from a microphone positioned in front of a loudspeaker 142 in a room 143 of a particular ambience. Responses (either linear using a single impulse test or non-linear as described in The Prior Application) can be obtained with the microphone at location 140 and at a different location 141 at a different distance from the loudspeaker, giving an amplitude reduction characteristic derived from moving away from the sound source. The two impulse responses or sets of them embody two different gains (as well as different ambience characteristics) and this pair of impulse response characteristics is now used in the method of simulation described herein to simulate compression and the compression effect simulated will be one of reducing dynamics by moving towards and away from the audio source according to amplitude of the source. This may be used for example to simulate the effect of a singer who has learned to moved to and from a microphone according to the volume of the voice to control dynamics.

A further example is illustrated in figure 12 in which a set of non linear impulse responses of a system 150 (which may also be stereo or other multi-channel system) is taken at progressively increasing levels comprising at least two different sets of levels 151 and 152 according to The Prior Application to generate response sets 153 and 154 where the system 150 exhibits gain reduction through progressively moving into non-linearity. Each FIR set is assigned a gain value for use in the gain reduction simulation according to an estimate of the gain of the system at the peak signal level, typically by estimating the peak deviation of the impulse response obtained from each sequence of test impulses. These are used as a set of non linear FIR sets of at least two intrinsic gains in the simulation method described herein to produce a compression effect characteristic of an overdriven system. System 150 may be a totally electronic device or for example an amplifier and speaker system with sound picked up by microphone.

By reference to figure 13 the exact operation of the bi-linear interpolation of figure 9 can be demonstrated.



FIR set storage 105 of figure 9 is also illustrated in figure 13 as a sequence of storage of sets of impulse responses. In this case each impulse response is shown with 8 elements for clarity but in practice many more elements will be used.

- 5 According to The Prior Application a factor  $k$  is determined to interpolate between two impulse responses in a FIR set (see for example figure 10 of The Prior Application). This is shown appearing in figure 13 where it is used to interpolate between elements of for example response #4 and #5 of FIRSET 1, and also between the matching elements of FIRSET 2. It is also shown in figure 9 at 120.

10

Interpolation factor  $j$  is also determined in equation (5) above as the interpolation between two sets of impulse responses representing two different gains. These are illustrated in figure 13 as FIRSET 1 and FIRSET 2.

- 15 In this way a single value 160 is derived from the 4 elements of the 4 impulse responses by bilinear interpolation using values  $j$  and  $k$  which are determined on a sample by sample basis for each step in the convolution process as the elements of each impulse response are stepped through in the convolution algorithm.

- 20 In practice, as described and shown in figure 14 of The Prior Application, it is convenient to generate part sums for each multiply accumulate operation for a given input sample, where  $j$ ,  $k$  and the pointers into the FIR set memory can be determined once for each sample and applied repeatedly. In this way for each input sample an output sample can be produced and part sums for subsequent output samples
- 25 assembled.

ADD  
21

21